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DESIGN OF SOURCE ARRAY BY INVERSE APPROACH

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Abstract

An inverse method was suggested for the design of source array to realize the desired sound field. The basic idea was stemmed from the acoustic holography as one of source identification techniques, which employs the measured field data to reconstruct the vibro-acoustic parameters on the source surface. In our problem, the pressure at specific points in the sound field was set as constraints for the desired sound field; surface points at the source plane became elementary sources. The whole procedure was conducted in three steps: First, the constraint for a desired sound field was set. The constraints could be either any desired field condition or measured data obtained in a target space. Two types of constraints could be classified: parameters related to sound quality and those related to the distributed characteristic information at a specified location. Second, the array source was modeled and feasibility test was conducted. Finally, the source parameters were calculated using the back propagation technique which has been used in the inverse source identification. Among many methods, the NAH based on the inverse boundary element method was adopted to obtain the source parameters. An example case of sound radiated from a piston was dealt with. Although the method is physically possible, the result revealed that a further study is needed to realize a complicated field from array sources.

INTRODUCTION

The target of sound field control is the generation of desired sound field for listeners. The characteristics of desired sound field depend on the purpose, situation, individual preference, and so on. Because there can be many types of desired field, the precise sound control method according to the target specification is required.

The loudspeaker array is frequently used for sound field control. Figure 1 shows the transfer path of signal between source and receiver. The input signal filter, $A_n(\omega)$, filters the input signal for each source. The subscript n is the index number of sources, and N denotes the total number of sources. The concept in sound control procedure is based on the change of transfer function between input and output signals by modifying the transfer function or adding other transfer functions. In many cases, the transfer function of an acoustic system, e.g., a room, is already determined and hard to modify; therefore, the sound field control problem reduced to the design of array response and input signal filter. In the design of array, however, many restrictions exist due to the limitation in specified target region and available number of sources. Consequently, the design of input signal filter is the most important and suitable for sound control.

Filtered input signal is what is supplied to each source as an input data. This means that the source strength of each array source is directly determined from $A_n(\omega)$. Therefore, once the desired shape of target sound field is given, the problem is very similar to the inverse source identification problem in general [1], in particular, the acoustic holography technique. The fundamental procedure of acoustic holography technique consists of three steps: measuring the sound pressure in the field, modelling the source or propagation path, reconstructing the surface pressure or velocity distribution on the source. In our sound source design problem, the field data in holography is replaced by the information on the desired field, and the source data in holography by the information of array source for generating the desired target field.

We have conducted the design procedure of loudspeaker array by inverse approach under the following three stages. The first stage was to set the constraints for a desired target sound field. The second was to model the array source. In some cases, the equation constructed for the desired sound filed did not have any solution that could be implemented by real situation. From that reason, a feasibility check was conducted first and, then, the required number of sources to describe the target field was determined as well as the elementary source size. The third stage was to obtain the source strength of each source by utilizing the holography method, analogously.

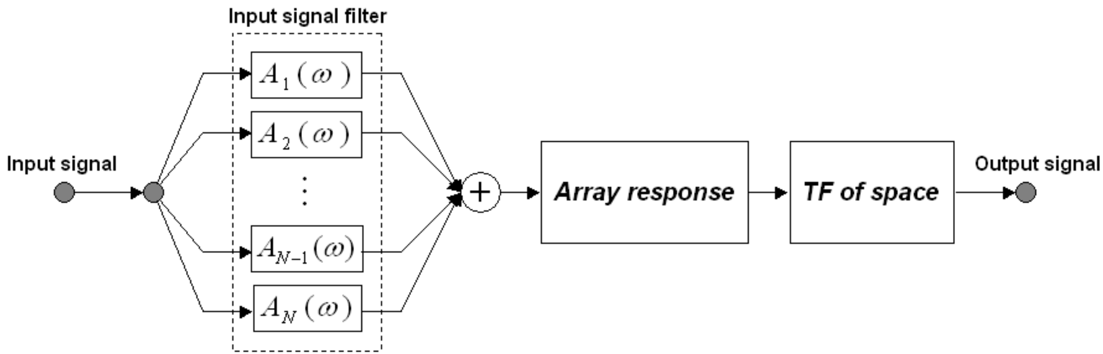

Figure 1 – Control of a sound field using a loudspeaker array system.

PREPARATION OF DESIRED FIELD DATA

In general, the description for the desired sound field is not clearly given. Two types of parameters could be thought of; one type is to express the perceptual feeling and the other type is related to the spatial distribution, thus objective data, in the field.

There exist various perceptual parameters and they are decided by the transfer function between input signal and output at a receiver point. It could be said that the same transfer function generates the same psychological feeling, if the input magnitude is maintained same. The transfer function between source and listening position can be written as [2]

$$H(\omega) = K \frac{(\omega-\omega_a)(\omega-\omega_b)\cdots}{(\omega-\omega_1)(\omega-\omega_2)\cdots}, \tag{1}$$

where ω_i ($i=a,b,c,...$) mean the zeros, ω_j ($j=1,2,3,...$) denotes the poles, and K is a constant of transfer function. By adding the input signal filter as shown in Fig. 1, the transfer function becomes

$$H'(\omega) = A(\omega)\cdot H(\omega). \tag{2}$$

Often, a flat or smooth frequency response, without any dip and sharp peak, is preferred although it is doubtful that the perceptual response corresponds to such frequency response. One should note that it is not easy to suggest a standard shape of transfer function for general purposes. Then, as an alternative way, measurement of the preferred sound field, e.g., the best seat in a renowned concert hall, can be used for simply quantifying the target. This method, however, needs a costly effort for measurement and survey of a field that matches with human preference.

The spatial distribution of sound can be described by relative SPL to the reference position. Three patterns can be the target field: focusing of sound to provide the acoustic information only to a specific range, distributing the sound in a uniform manner to transfer the same information to the whole selected range, and dividing the sound to provide different information into each selected region. In any case, the definition of control range is important for the array system and field condition because they are somehow limited.

BASIC THEORY OF NAH TECHNIQUE [3-5]

In this study, the near-field acoustic holography (NAH) technique based on the boundary element method (BEM) is adopted to find the source strength to satisfy the target sound field. Starting from the well-known Kirchhoff-Helmholtz integral equation for harmonic acoustic field, one can obtain the following matrix-vector relation for the radiated sound field:

$$\boldsymbol{p}_f = (\boldsymbol{D}_f \boldsymbol{D}_s^{-1} \boldsymbol{M}_s + \boldsymbol{M}_f)\boldsymbol{v}_s \equiv \boldsymbol{G}_v \boldsymbol{v}_s, \text{ or } \boldsymbol{p}_f = (\boldsymbol{D}_f + \boldsymbol{M}_f \boldsymbol{M}_S^{-1} \boldsymbol{D}_S)\boldsymbol{v}_s \equiv \boldsymbol{G}_p \boldsymbol{p}_s \tag{3}$$

Here, $\boldsymbol{p}_s$ and $\boldsymbol{v}_s$ mean the pressure and velocity of surface, $\boldsymbol{p}_f$ and $\boldsymbol{v}_f$ are the pressure and velocity of field, $\boldsymbol{D}_s$ and $\boldsymbol{M}_s$ denote the monopole and dipole matrix on the surface, and $\boldsymbol{D}_f$ and $\boldsymbol{M}_f$ mean the matrix of field, respectively. Equation (3) is the standard form of BEM for solving the forward problems.

For the backward or inverse problem, the surface velocity can be obtained by the least-squared solution and the singular value decomposition of transfer matrix $\boldsymbol{G}$ as

$$\hat{\boldsymbol{v}}_s = (\boldsymbol{G}_v^H \boldsymbol{G}_v)^{-1} \boldsymbol{G}_v^H \boldsymbol{p}_f \equiv (\boldsymbol{G}_v)^+ \boldsymbol{p}_f = \boldsymbol{W}_v \Lambda_v^{-1} \boldsymbol{U}_v^H \boldsymbol{p}_f, \tag{4}$$

where the operator H signifies the Hermitian operator, and the superscript $^+$ means the pseudo-inverse matrix form, the diagonal elements λ_i of the matrix Λ represent the singular values of the matrix $\boldsymbol{G}_v$, and $\boldsymbol{U}$, $\boldsymbol{W}$ are the unitary singular matrices.

DEMONSTRATION EXAMPLE

Preparation of target field data

Consider the sound field generated by a circular piston as given by [6]

$$p(r,\theta,t) = \frac{j}{2}\rho_0 c U_0 \frac{a}{r} ka \left[\frac{2J_1(ka\sin\theta)}{ka\sin\theta}\right] e^{j(\omega t - kr)}, \tag{5}$$

where ρ_0 is the density of air, c the speed of sound, a the radius of piston, and U_0 the velocity magnitude of source. If the radius of piston changes, the radiation pattern also changes at the same frequency. In this example, the sound field generated by circular piston (1 kHz), varying the radius as 0.1, 0.2, 0.4, and 0.8 m, was set as target field.

Array source modelling using BEM

We considered an array system consisting 15 elementary sources (loudspeakers) with equal spacing. Each elementary source was model by 5 nodes and 4 triangular elements. Nodes at the corner represented fixed boundaries. Each elementary source was modelled as an omni-directional source to neglect the size effect. Figure 2 shows the directivity pattern of a single piston source as a function of Helmholtz number, kd, where k is the wave number and d is the aperture size of the source. For $kd << 1$, the omni-directional source assumption was valid within 10% error. To satisfy this condition, the characteristic length of BEM model was determined as 3 mm (*kd=0.1*), which limits the applicable high frequency to about 20 kHz, considering $\lambda / 6$ criterion. The spacing between sources was twice the characteristic length of an element.

Result

In Figs. 3(a), (c), (e) and (g), the predicted sound level distribution can be seen varying the piston radius as 0.1, 0.2, 0.4, and 0.8 m (at 1 kHz). In Figs. 3(b), (d), (f), and (h), one can see the generated field by sources with the source strength derived by inverse-BEM. The difference between two fields increases as the field became more complicated (Table 1). Especially, note that sharp dips were almost smoothed. This suggests that it is not easy to generate a dramatically changing field by using this technique.

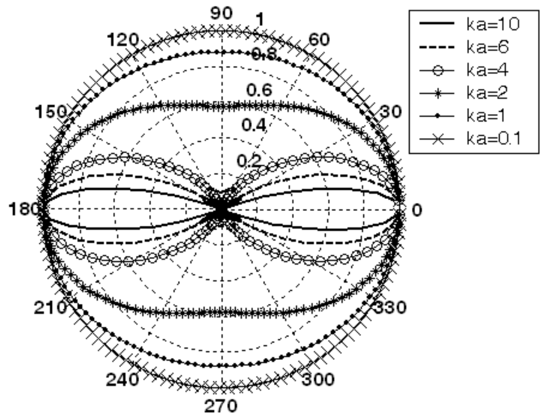

Figure 2 – Directivity pattern of single circular piston source.

SUMMARY

Design process of a source array was suggested in order to obtain the desired sound field. The whole procedure was conducted by three stages: First, the constraint for a desired sound field was set. Second, the array source was modeled and feasibility test was conducted. Finally, the source parameters were calculated using the back propagation technique which has been used in the NAH based on the inverse BEM. An example case of sound radiated from a piston was dealt with. A gradual change of target field could be realized with small difference, the result revealed that a further study is needed to realize a complicated field from array sources.

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Table 1 – Difference between target field calculated by theory and inversely generated field.

Radius of piston (m)	0.1	0.2	0.4	0.8
Difference reference to target field (%)	6.9	11.5	21.9	30.5

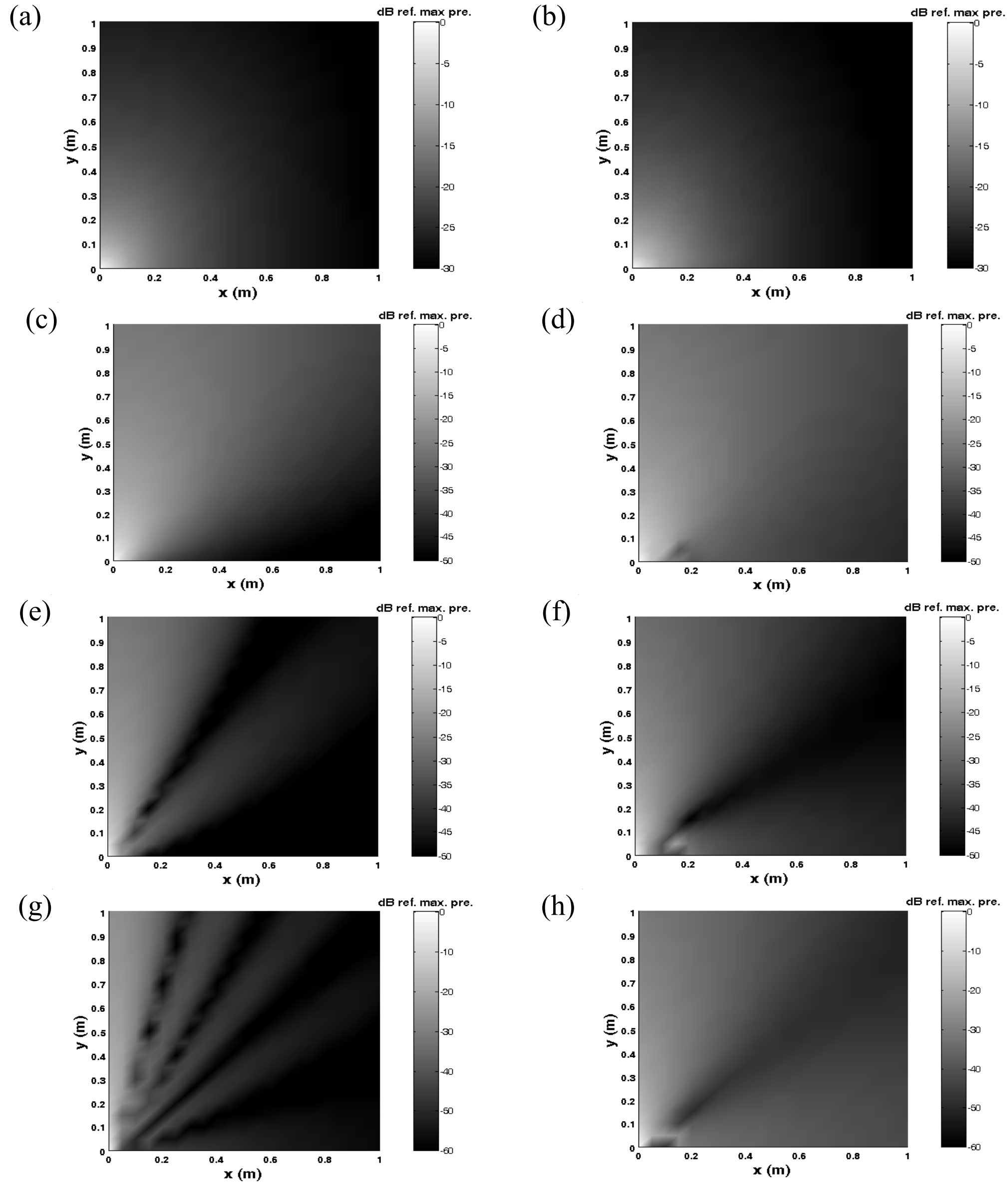

Figure 3 – Comparison of sound level distribution of target field: theory vs. inverse calculation. (a),(b) target field with a=0.1; (c),(d) a=0.2; (e),(f) a=0.4; (g),(h) a=0.8. (a),(c),(e),(g) theoretical field (target field); (b),(d),(f),(h) inversely generated field from target field.